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NOVEL DIGITAL SIGNAL PROCESSING AND DETECTION TECHNIQUES.(U)
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FINAL SCIENTIFIC REPORT

Grant [redacted]-AFOSR 76-3083

NOVEL DIGITAL SIGNAL PROCESSING AND DETECTION TECHNIQUES

(1 September 1976 - 30 August 1981)

Submitted by:

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FINAL SCIENTIFIC REPORT

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NOVEL DIGITAL SIGNAL PROCESSING AND DETECTION TECHNIQUES

(1 September 1986 - 30 August 1981)

This report summarizes the research conducted under Grant [redacted]-AFOSR 76-3083 during the period from 1 September 1976 to 30 August 1981

In the area of narrowband signal processing, design rules are developed for optimum decimator and interpolator, a new efficient scheme using recursive filter for decimation/interpolation is proposed, and a novel approach to the computation of narrowband spectra is shown to yield substantial saving over conventional approaches. Results on the implementation of recursive filters with poles near the unit circle that produces significantly reduced roundoff error include a transformation technique, a scheme to modify the quantizer error spectrum, and a new computationally efficient low noise filter structure. In the area of nonclassical signal detection, several results were derived on nonparametric sequential procedures and on the quantization of signal for detection. In addition, a programmable charge transfer device filter is developed, several problems concerning ADPCM are investigated, results are obtained on FFT roundoff error including the prime factor algorithm, and an effective method of generating random sequences is studied.

The results of research supported by this Grant are summarized below. The number in the brackets refers to items in Appendix A which contains a list of publications supported by the Grant. Appendix B lists the Ph.D. dissertations supported by the Grant.

I. Narrowband Signal Processing

It often occurs in practice that a signal of interest occupies only a narrowband in the spectrum of an available data sequence. Examples can be found in communication systems, radar, sonar and other surveillance systems. Standard digital signal processing techniques when applied to these narrowband signals require substantial storage and high computation rate as well as long word length in the implementation. These difficulties can be circumvented by using multirate processing, in which the sampleing rate is changed along the signal path. We summarize below a number of results on this topic obtained during the Grant period.

I.1 Narrowband Filtering

The performance of multirate filters is analyzed to show that the error (when compared with an ideal filter) consists of two components, one due to frequency aliasing and the other due to the deviation of te actual filter characteristics from the ideal [6,12]. Design rules are then developed for selecting parameters for optimally designing the filter [12]. In the case of band-pass filtering, it is shown that the factor by which the sampling rate is changed may not be arbitrarily chosen, and a necessary and sufficient condition on the "admissibility" of the factor is derived [18].

The basic building block in multirate processing is the decimation and the interpolation, by which the sampling rate

is decreased and increased respectively. Traditionally, non-recursive (FIR) filters are used in decimation/interpolation because a computation rate reduction is preceived not to be possible with recursive (IIR) filters. However, by casting the decimation/interpolation as periodically time varying processes, it is possible to arrive at a scheme of using recursive filters with similar computational advantages. This leads to a multirate processors that are substantially more efficient by as much as an order of magnitude than the previously ones using nonrecursive filters [38].

I.2 Narrowband Spectrum Analysis

The frequency spectrum of a long data sequence over a narrow band can be efficiently computed by decimating the sequence using FIR and calculating an FFT on the shortened sequence. This method offers a low computation rate, a small storage requirement, good frequency resolution, and a controllable accuracy [17]. It is shown that, in the limit, no more than 4 multiplications are needed for each input data. Cooley and Winograd showed that the number of computation can be further reduced for the lowpass case by relaxing the filter constraints. This improvement is extended to the general bandpass case [31]. An algorithm is developed for choosing the best decimation factor and designing appropriate decimating filters for any chosen frequency band. In the limit, the number of multiplications is less than 3 and may be as low as 1.25 per input sample [31].

The approach described above makes use of an estimate of the order of FIR filter of the bandpass and bandstop type. A simple estimate was developed that is accurate to within $\pm 4\%$ in the majority of cases and to within $\pm 6\%$ in over 95% of the cases tested [14,20].

II. Implementation of Digital Filters

To implement a digital filter that realizes a given transfer function, a veriety of approaches are available. Special attention must be given, however, when the filter is of the recursive type and the poles are close to the unit circle. Foundoff error and coefficient sensitivity can be very large if these high Q filters are implemented in a straightforward manner. Indeed, it can be shown that by using B bit words the mean squared roundoff error is of the order $s^{-2B}\ \delta^{-1}(\delta^2+\theta^2)^{-1}$ for poles at $(1-\delta)e^{\pm j\theta}$. We summarize below the results on low noise implementation using three distinct approaches.

II.1 Low Noise Structure for High Q Poles

Agarwal and Burrus have propsed tow all-pole digital filter structures which exhibit small error for high Q poles. These results are extended to include zeros and for the ROM/ Accumulator realization. It is shown that for these low noise strucutres, the error is of the form $a\cdot 2^{-2B}[b\delta^{-1}+c+d\theta^4\delta^{-1}(\theta^2+\delta^2)^{-1}]$ where a,b,c, and d are constants [15,16,23,24]. This error is seen to be substantially smaller than that for the direct implementation. The difference is usually in the order of 3 to 7 bits. Thus the new structures can offer considerable advantage in terms of hardware complexity and operating speed.

II.2 Error Spectrum Shaping Quantizer

Although the roundoff noise introduced at a quantizer has essentially flat spectrum, the "in band" component can sometimes be amplified to a greater extent than the signal itself if high Q poles are present. By feeding back the local error generated by the quantizer, it is possible to significantly reduce the error build up. This approach can be shown to be equivalent to modifying the spectrum of the quantizing error, hence the name error spectrum shaping quantizer (ESSQ) [5]. The employment of ESSQ not only reduces the output quantization error, it can also significantly reduce, or even eliminate, zero input limit cycle oscillation [19,30]. It is further shown that if one extra multiplication is used, then the amplitude of these oscillations can not exceed 2 bits [37].

II.3 Low Noise Computationally Efficient Filter Structures

To meet given amplitude specifications elliptic filters are known to be optimal among recursive filters of a given order. Conventional cascade realizations of elliptic filters of order N require approximately (3N/2) multiplications per output sample. We have developed a new recursive filter structure which requires only about two-thirds as many multiplications as are required in conventional realizations of elliptic filters to meet the same amplitude specifications [32]. The analytic optimum solution to the problem of design of odd order filters can be shown to be elliptic filter itself. There is no such correspondence for even order filters. In addition to the saving

in computations the scheme also lends itself to realizations with minimal quantization error.

III. Non-Classical Signal Detection

Classical hypothesis testing procedures produce optimal signal detectors when fixed sample sizes are used and when a complete parametric statistical characterization of the input observation is available. However, it often occurs in practice that a complete statistical description is not available. Furthermore, it is well known that fixed sample size techniques are not as efficient as sequential procedures. We summarize below the results of our investigation on sequential detection and on quantization for detection.

III.1 Sequential Detection

A class of nonparametric sequential tests for testing a symmetric density hypothesis against a one sided shift alternative is investigated [2]. This proposed procedure requires less time for the ranking procedure and is more suitable for real time implementation. The test statistic is the sum of intermediate statistics obtained from the ranks within the most recent m observations [21]. Approximte expressions for the power and average sample number are derived, and the effectiveness of this approach is demonstrated [35]. It is shown that a properly truncated sequential probability ratio test can significantly reduce the average sample number when the parameter values of the sample distribution lie between those for the hypothesis and for the alternative [9]. The asymptotic behavior of the relative

efficiency of the sequential probability ratio test is studied for the detection of a constant signal in additive noise. Results are obtained for the asymptotic behavior of the realtive efficiency of the sequential probability ratio test for the case of constant signal in additive noise [11].

III.2 Quantization for Signal Detection

The sequential dead-zone limiter detector originally proposed by Shin and Kassam may be treated as a generalized ruin problem that includes ties, and the sequential four level sign detector may be treated as another generalization of the classical ruin problem [22]. These two detectors are analyzed using this approach and their performances compared to that of the more familiar sequential sign detector in terms of relative efficiency and asymptotic relative efficiency [36].

The Ali-Silvy distance measures is applied to the problem of designing quantizers for use in binary detection systems [8]. Necessary condtions are established for the selection of optimum quantizer parameters by using a slight generalization of the notion of quantization. Agreement with the asymptotic results based on Pitman efficiency is established for small-signal conditions. Necessary conditions are also establsihed for the optimum selection of quantizer breakpoints for signal detection systems operating in m-dependent noise environment [13].

IV. Other Problems in Digital Signal Processing

IV.1 Programmable Charge Transfer Device Filters

Charge transer devices (CTD) play an important role in sampled analogue signal processing, particularly in applications where hgih-order filtering is needed. A severe drawback of these filters is that the filter coefficients are fixed by the metalisation pattern on the semiconductor, and cannot be changed by the user.

A novel filtering scheme based on delta modulation of the filter coefficient sequence is developed [27,28]. The filter employs only coefficients of +1, -1, and 0. These coefficients can be implemented by tapping the electrode in a CCD filter through a programmable CCD filter. It is shown that arbitrary low-pass filter characteristics can be realized by using this approach [29]. This approach is then extended to highpass and bandpas filters [33]. Various techniques to improve upon the performance are disclosed [33].

IV.2 Adaptive Differential Pusle Code Modulation (ADPCM)

ADPCM is an attractive digital coding technique whose value in digital signal processing has been well investigated. We have perviously developed nonrecursive filters using ADPCM. A technique for computing the power spectra of signals modulated by ADPCM without the significant use of hardware multipliers is developed [1]. A method is devised to compute the marginal quantization error and step size distribution in an ADPCM system

with a number of different Markov input signals [4]. The problem of stability of an ADPCM system is investigated for both deterministic and stochastic points of view [7].

IV.3 Fast Fourier Transform (FFT) Error

Statistical models for roundoff error are used to predict the computation error in fast Fourier transfrom (FFT) algorithms [3]. In addition to the radix 2 case, some results are also derived for the arbitrary radix, the mixed radix, and the multi-dimensional cases. The prime factor FFT makes use of some recent computational complexity results by Winograd to compute the DFT with a fewer number of multiplications than required by the FFT. An expression for the MSE due to roundoff error when floating point arithmetic is used is derived and a simple bound on the MSE is obtained [26,34].

IV.4 Generation of Random Sequence

Simulation in communications, radar, sonar, speech, and other disciplines are often conducted based upon a probabilistic system description. Althoughthe parameterization of any stochastic process in the system is usually incomplete, two statistics often available are the first order probability distribution and the autocovariance function. It is easy to generate a random sequence having either a specificed marginal distribution or a specified autocovariance. However, for other than the Gaussian case, the joint problem has received little attention, even though

processes arising in the above disciplines are often both correlated and non-Gaussian.

One scheme to generate such a sequence is a white Gaussian input to a digital filter followed by a zero memory nonlinearity. The nonlinearity is chosen so that the desired probability distribution is realized exactly and the digital filter is designed so that the desired autocovariance function is approximated. A number of theoretical results concerning this scheme is derived [25].

APPENDIX A

Publication Supported by Grant AF AFOSR 76-3083

1. "Power Spectra of ADPCM," B. Liu and L.H. Goldstein, IEEE Trans. Acoustics, Speech, and Signal Processing, Vol. ASSP-25 February 1977, pp. 56-62.

2. "A Modifeid Sequential Rank Test for Real-Time Implementation," S. Tantaratana and J.B. Thomas, Proceedings of the 1977 Conference on Information Sciences and Systems, Johns Hopkins University, Baltimore, Md., March 30-April 1, 1977, pp. 395-400.

3. "Accumulation of Roundoff Error in Floating Point FFT," T. Thong and B. Liu, IEEE Trans. Circuits and Systems, Vol. CAS-24, March 1977, pp. 132-143.

4. "Quantization Error and Step Size Distribution in ADPCM," L.H. Goldstein and B. Liu, IEEE Trans. Information Theory, Vol. IT-23, April 1977, pp. 216-223.

5. "Error Spectrum Shaping in Narrow-Band Recursive Filters," T. Thong and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing, April 1977, pp. 200-203.

6. "Aliasing Error in the Multirate Implementation of Narrow-band Filters," F. Mintzer and B. Liu, 1977 International Conference on Acoustics, Speech, and Signal Processing, May 1977, pp. 105-108.

7. "Deterministic and Stochastic Stability of Adaptive Differential Pulse Code Modulation," L.H. Goldstein and B. Liu, IEEE Trans. on Information Theory, Vol. IT-23, No. 4, July 1977, pp. 445-453.

8. "Applications of Ali-Silvery Distance Measrues in the Design of Generalized Quantizers for Binary Decision Systems," H.V. Poor and J.B. Thomas, IEEE Trans. on Communications, Vol. COM-25, No. 9, September 1977, pp. 893-900.

9. "Truncated Sequential Probability Ratio Test," S. Tantaratana and J.B. Thomas, Information Sciences, Vol. 13, No. 3, November 1977, pp. 283-300.

10. "Bivariate Densities with Diagonal Expansions in Gegenbauer Polynomials," H. Derin, G.L. Wise and J.B. Thomas, Journal of the Franklin Institute, Vol. 304, No. 6, December 1977, pp. 243-249.

11. "Relative Efficiency of Sequential Probability Ratio Test in Signal Detection,: J.B. Thomas and S. Tantaratana, IEEE Trans. Information Theory, Vol. IT-24, No. 1. January 1978, pp. 22-31.

12. "Aliasing Error in the Design of Multirate Filters," F. Mintzer and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing, Vol. ASSP-26, No. 1, February 1978, pp. 76-88.

13. "Optimum Quantization for Memoryless Detection in m-Dependent Noise," H.V. Poor and J.B. Thomas, Proceedings of the 1978 Conference on Information Sciences and Systems, Johns Hopkins University, Baltimore, Md., March 29-31, 1978, pp.

14. "An Estimate of the Order of An Optimum FIR Bandpass Digital Filter," F. Mintzer and B. Liu, IEEE 1978 International Conference on Acoustics, Speech, and Signal Processing, April 1978, pp. 483-486.

15. "ROM Realization of Digtial Filters for Poles Near the Unit Circle," D. Munson and B. Liu, Proceedings of 1978 IEEE International Symposium on Circuits and Systems, May 1978, pp. 999-1003.

16. "Low-Noise Realizations for Digital Filters with Poles Near the Unit Circle," D.C. Munson, Jr. and B. Liu, Sixteenth Annual Allerton Conference on Communication, Control, and Computing, October 4-6, 1978, pp. 372-381.

17. "Calculation of Narrow-Band Spectra by Direct Decimation," B. Liu and F. Mintzer, IEEE Trans. on Acoustics, Speech, and Signal Processing, Vol. ASSP-26, No. 6, December 1978, pp. 529-534.

18. "The Design of Optimal Multirate Bandpass and Bandstop Filters," F. Mintzer and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing, Vol. ASSP-26, No. 6, December 1978, pp. 534-543.

19. "Narrowband Recursive Filters with Error Spectrum Shaping, D.C. Munson and B. Liu, IEEE 1979 International Conference on Acoustics, Speech and Signal Processing, April 1979, pp. 367-370.

20. "Practical Design Rules for Optimum FIR Bandpass Digital Filters," F. Mintzer and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing, Vol. ASSP-27, April 1979, pp. 204-206.

21. "On Ranked Sequential Tests," S. Tantaratana and J.B. Thomas, Proceedings of the 22nd Midwest Symposium on Circuits and Systems, Philadelphia, Pa., June 17-19, 1979, pp. 519-524.

22. "Generalizations of the Sequential Sign Detector," C.C. Lee and J.B. Thomas, Proc. Seventeenth Annual Allerton Conf. on Comm. Control, and Computing, October 1979, pp. 848-857.

23. "ROC/ACC Realizations of Digital Filters for Poles Near the Unit Circle," D.C. Munson Jr. and B. Liu, IEEE Trans. on Circuits & Systems, Vol. CAS-27, No. 2, February 1980, pp. 147-151.

24. "Low-Noise Realizations for Narrow-Band Recursive Digital Filters," D.C. Munson Jr, and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing, Vol. ASSP-28, No. 1, February 1980, pp. 41-54.

25. "On Computer Generation of Random Sequences," D.C. Munson and B. Liu, Proc. Fourteenth Annual Conf. on Info. Sciences, and Systems, Princeton, NJ, March 1980.

26. "Floating Point Error Bound in the Prime Factor FFT," D.C. Munson Jr., and B. Liu, IEEE Int. Conf. on Acoustics, Speech and Signal Processing, April 1980, pp. 69-72.

27. "Programmable CTD Fitlering Uisng Coefficients 0, +1, -1," M.R. Bateman and B. Liu, IEEE Inst. Sym., Circuits and Systems, April 1930, pp. 134-137.

28. "An Approach to Programmable CTD Filters Using Coefficients 0, +1, and -1," M.R. Bateman and B. Liu, IEEE Trans. on Circuits and Systems, Vol. CAS- 27, No. 6, June 1980, pp. 451-456.

29. "Signal Detection Based on Discrete Data," C.C. Lee and J.B. Thomas, Eighteenth Annual Allerton Conf. on Comm., Control and Computing, October 1980, pp.

30. "Narrowband Recursive Filters with Error Spectrum Shaping," D.C. Munson, Jr. and B. Liu, IEEE Trans. on Circuits and Systems, Vol. CAS-28, No. 2, February 1981, pp. 160-163.

31. "On Narrow-Band Spectrum Calculation by Direct Decimation," M. Quirk and B. Liu, IEEE 1981 Int. Conf. Acoustics, Speech, and Signal Processing, April 1981, pp. 85-88.

32. "A Class of Low Noise Computationally Efficient Recursive Digital Filters," R. Ansari and B. Liu, IEEE 1981 Int. Symposium on Circuits and Systems, April 1981, pp. 550-553.

33. "Programmable CTD Filtering Using Coefficients 0, +1, and -1," M.R. Bateman and B. Liu, IEE Proc. Vol. 128, Pt. G, No. 4, August 1981, pp. 208-212.

34. "Floating Point Roundoff Error in the Prime Factor FFT," D.C. Munson, Jr. and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing, Vol. ASSP-29, No. 4, August 1981, pp. 877-882.

Papers Accepted for Publication

35. "A Class of Nonparameteric Sequential Tests," S. Tantarantana and J.B. Thomas, IEEE Trans. on Info. Theory.

36. "Sequential Detection Based on Simple Quantization," C.C. Lee and J.B. Thomas, Journal of Franklin Institute.

37. "Limit Cycle Bounds for Digital Filters with Error Spectrum Shaping," M.R. Bateman and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing.

38. "Efficient Sampling Rate Alteration Using Recursive (IIR) Digital Filters," R. Ansari and B. Liu, IEEE Trans. on Acoustics, Speech, and Signal Processing.

APPENDIX B

Ph.D. Dissertations Supported Fully and Partially by Grant AF AFOSR 76-3083

1. M. Bateman, "Some Novel Techniques for Digital and Sampled Processing," 1980, (Now with Hewlett Packard, Palo Alto, CA)

2. G. Cybenko, "Error Analysis of Some Signal Processing Algorithms," 1978, (Now with Dept. of Mathematics, Tufts Univ., Medford, MA)

3. C.C. Lee, "On Nonparametric, Sequential & Mixed-Sampled-Size Signal Detection Procedures," 1980, (Now with Dept. of Electrical Engineering, Northwestern University, Evanston, IL)

4. F.J. Mintzer, "Narrowband Digital Signal Processing," 1978, (Now with IBM Research, Yorktown, NY)

5. D.C. Munson, Jr., "Some New Techniques and Performance Analyses in Digital Signal Processing," 1979, (Now with Dept. of Electrical Engineering, University of Illinois, Urbana, IL)

6. H.V. Poor, "Topics in Optimal and Robust Detection - Quantization, Stochastic-Signal, and Memoryless Detection," 1977, (Now with Dept. of Electrical Engineering, University of Illinois, Urbana, IL)

7. S. Tantaratana, "On the Relative Efficiencies of Some Parametric and Non-parametric Sequential Detectors," 1977, (Now with Dept. of Electrical Engineering, Auburn University, Auburn, AL)

8. M.D. Wood, "Signal Detection in Correlated Noise," 1978, (Now with TRW, Los Angeles, CA)

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